AMENDMENTS TO THE CLAIMS

Please amend Claims 10, 12, 19, 21, 22, 24, 27, 39, 56, 58 and 59 as follows:

1. (Previously presented) A method of separating a desired speech signal in an acoustic environment, comprising:

receiving a plurality of input signals, the input signals being generated responsive to the desired speech signal and other acoustic signals;

processing the received input signals using an independent component analysis (ICA) or blind source separation (BSS) method under stability constraints, wherein the ICA or BSS method modulates the mathematical formulation of mutual information directly or indirectly through approximations; and

separating the received input signals into output channels comprising one or more desired audio output signals and one or more noise output signals.

2. (Original) The method according to claim 1, wherein one of the desired audio signals is the desired speech signal.

3. (Previously presented) The method according to claim 1, wherein the ICA method further comprises minimizing or maximizing the mathematical formulation of mutual information directly or indirectly through approximations.

4. (Previously presented) The method according to claim 1, wherein the stability constraints comprise pacing the adapting of an ICA filter.

5. (Previously presented) The method according to claim 1, wherein the stability constraints comprise scaling the received input signals using an adaptive scaling factor, the adaptive scaling factor being selected to constrain weight adaptation speed.

6. (Previously presented) The method according to claim 1, wherein the stability constraints comprise filtering learned filter weights in the time domain and the frequency domain, the filtering selected to avoid introduction of artificial reverberation effects.

7. (Previously presented) The method according to claim 1, further comprising applying peripheral pre-processing or post-processing techniques to at least one of the received input signals or at least one of the separated output signals.

8. (Previously presented) The method according to claim 1, further comprising pre-processing the received input signals.

9. (Original) The method according to claim 8, further comprising improving the conditioning of a mixing scenario applied to the input signals.

10. (Currently amended) The method according to claim 2, further comprising utilizing characteristic information of the desired speech signal to identify the output channel ~~containing~~ comprising the separated desired speech signal.

11. (Original) The method according to claim 10 wherein the characteristic information is spatial, spectral or temporal information.

12. (Currently amended) The method according to claim 1, further comprising applying post-processing techniques to at least one of the ~~separated~~ output signals using at least one processing signal selected from one or more of the noise signals and one or more of the input signals.

13. (Previously presented) The method according to claim 12, wherein the using at least one processing signal consists of using the noise signal.

14. (Previously presented) The method according to claim 13 wherein the using the noise signal comprises using the noise signal to estimate the noise spectrum for a noise filter.

15. (Previously presented), The method according to claim 1, further comprising:

spacing apart at least a first and a second microphone; and

generating one of the input signals at each respective microphone.

16. (Previously presented) The method according to claim 15, wherein the spacing apart at least a first and a second microphone comprises spacing the microphones between about 1 millimeter and about 1 meter apart.

17. (Previously presented) The method according to claim 15, wherein the spacing apart at least a first and a second microphone comprises spacing the microphones apart on a telephone receiver, a headset, or a hands-free kit.

18. (Previously presented) The method according to claim 1, wherein the ICA or BSS method comprises:

adapting a first adaptive ICA filter connected to a first output signal and to a second input signal by a recursive learning rule involving the application of a nonlinear bounded sign function to one or more noise output signals; and

adapting a second adaptive ICA filter connected to a first input signal and to a second output signal by a recursive learning rule involving the application of a nonlinear bounded sign function to the one or more desired audio output signals,

wherein the first filter and the second filter are repeatedly applied to produce the desired speech signal.

19. (Currently amended) The method according to claim 18, further comprising:

spacing apart at least a first and a second microphone;

generating one of the input signals at each respective microphone;

recursively filtering the one or more desired audio output signals by the first adaptive independent component analysis filter to obtain a recursively filtered speech signal;

recursively filtering the one or more noise output signals by the second adaptive independent component analysis filter to obtain a recursively filtered noise signal;

adding the recursively filtered speech signal to the input signal from the second microphone, thereby producing the <u>one or more noise output signals</u> ~~noise output signal~~; and

adding the recursively filtered noise signal to the input signal from the first microphone, thereby producing the one or more desired audio output signals.

20. (Previously presented) The method according to claim 19, wherein the received input signals are inversely scaled by an adaptive scaling factor computed from a recursive equation as a function of incoming signal energy.

21. (Currently amended) The method according to <u>claim 18</u> ~~claim 1~~, further comprising:

stabilizing a recursive learning rule adapting the first adaptive ICA filter by smoothing coefficients of the first adaptive ICA filter in time; and

stabilizing a recursive learning rule adapting the second adaptive ICA filter by smoothing coefficients of the second adaptive ICA filter in time.

22. (Currently amended) The method according to <u>claim 18</u> ~~claim 1~~, wherein the filter weights of the first adaptive ICA cross are filtered in the frequency domain, and wherein the filter weights of the second adaptive ICA cross are filtered in the frequency domain.

23. (Previously presented) The method according to claim 1, further comprising post

processing the desired speech signal comprising voice activity detection and wherein post-processed outputs are not fed back to input signals.

24. (Currently amended) The method according to claim 18 ~~claim 1~~, wherein the ICA method is implemented in a fixed point computing precision environment and wherein the ICA method further comprises:

applying the adaptive ICA filters at every sampling instant;

updating filter coefficients at multiples of the sampling instant; and

adapting filter lengths of variable sizes according to the computational power available.

25. (Previously presented) The method according to claim 1, further comprising applying spectral subtracting to the one or more desired audio output signals based on the one or more noise signals.

26. (Previously presented) The method according to claim 1, further comprising applying Wiener filtering to the one or more desired audio output signals based on the one or more noise signals.

27. (Currently amended) The method according to claim 1, further generating a third set of audio input signals ~~from~~ at a third microphone, and applying a nonlinear bounded function to incoming signals using a third filter.

28. (Canceled)

29. (Canceled)

30. (Canceled)

31. (Canceled)

32. (Canceled)

33. (Canceled)

34. (Canceled)

35. (Canceled)

36. (Canceled)

37. (Canceled)

38. (Canceled)

39. (Currently amended) A system for separating a desired speech signal in an acoustic environment, comprising

a plurality of input channels each receiving one or more acoustic signals, wherein the one or more acoustic signals comprises a speech signal;

at least one independent component analysis (ICA) or blind-source separation (BSS) filter module comprising an ICA or BSS filter that separates the received signals into one or more desired audio signals and one or more noise signals;

a stability constraint, wherein the stability constraint at least partially stabilizes the ~~at least one~~ ICA or BSS filter; and

a plurality of output channels transmitting the separated signals,

wherein the filter modulates the mathematical formulation of mutual information directly or indirectly through approximations.

40. (Previously presented) The system according to claim 39, wherein the one or more acoustic signals comprise the one or more desired audio signals.

41. (Canceled)

42. (Previously presented) The system according to claim 39, wherein implementing the stability constraint paces adaptation of the ICA or BSS filter.

43. (Previously presented) The system according to claim 39, wherein implementing the stability constraint comprises scaling ICA or BSS inputs using an adaptive scaling factor, the adaptive scaling factor selected to constrain adaptation speed.

44. (Previously presented) The system according to claim 39, wherein implementing the stability constraint comprises filtering learned filter weights in the time domain and the frequency domain, the filter selected to avoid introduction of artificial reverberation effects.

45. (Previously presented) The system according to claim 39, further comprising one or more processing modules comprising at least one filter selected from a pre-processing peripheral filter and a post-processing peripheral filter applied to the one or more acoustic signals and/or the separated signals.

46. (Previously presented) The system according to claim 45, wherein the filter is the pre-processing peripheral filter.

47. (Previously presented) The system according to claim 45, wherein the filter is the post-processing peripheral filter.

48. (Previously presented) The system according to claim 39, further comprising one or more microphones connected to the plurality of input channels.

49. (Previously presented) The system according to claim 48, wherein the one or more microphones are two or more microphones, and wherein each of the two or more microphones is spaced between about 1 millimeter and about 1 meter apart.

50. (Original) The system according to claim 39, wherein the system is constructed on a hand-held device.

51. (Previously presented) The system according to claim 39, wherein the at least one ICA or BSS filter module comprises:

a first adaptive independent component analysis (ICA) filter connected to a first output channel and to a second input channel, the first filter being adapted by a recursive learning rule involving the application of a nonlinear bounded sign function to the one or more noise signals;

a second adaptive independent component analysis filter connected to a first output channel and to a second input channel, the second filter being adapted by a recursive learning rule involving the application of a nonlinear bounded sign function to the desired speech signal;

wherein the first filter and the second filter are repeatedly applied to produce the desired speech signal.

52. (Canceled)

53. (Canceled)

54. (Canceled)

55. (Previously presented) The system according to claim 39, wherein the plurality of input channels comprises at least two spaced-apart microphones constructed to receive the acoustic signals, the microphones being an expected distance from a speech source;

wherein the at least one ICA or BSS filter module is coupled to the microphones; and

wherein the at least one ICA or BSS filter module is configured to:

receive sound signals from the two microphones; and

separate the sound signals under the stability constraint into at least one desired speech output signal line and at least one noise output signal line.

56. (Currently amended) The system according to claim 55, further comprising a post-processing filter coupled to the noise output signal line and to the desired speech output signal line.

57. (Previously presented) The system according to claim 55, wherein the microphones are spaced about 1 millimeter to about 1 meter apart.

58. (Currently amended) The system according to claim 57 further comprising a pre-processing module configured to pre-process the acoustic ~~sound~~ signals received at each microphone.

59. (Currently amended) The system according to claim 55, wherein one of the microphones is on a face of a device housing and another of the microphones ~~the other microphone~~ is on another face of the device housing.

60. (Previously presented) The system according to claim 55, wherein the system is integrated into a speech device.

61. (Previously presented) The system according to claim 60, wherein the speech device comprises a wireless phone.

62. (Previously presented) The system according to claim 60, wherein the speech device comprises a hands-free car kit.

63. (Previously presented) The system according to claim 60, wherein the speech device comprises a headset.

64. (Previously presented) The system according to claim 60, wherein the speech device comprises a personal data assistant.

65. (Previously presented) The system according to claim 60, wherein the speech device comprises a handheld bar-code scanning device.